IN THE CLAIMS

The following is a complete listing of revised claims with a status identifier in parenthesis.

LISTING OF CLAIMS

1. (Original) A system for processing audio signals comprising:

(a) a splitter for dividing an input audio signal into a first and one or more secondary signal portions, which in combination provide a complete representation of the input signal, wherein the first signal portion contains information sufficient to reconstruct a representation of the input signal;

(b) a first encoder for providing encoded data about the first signal portion, and one or more secondary encoders for encoding said secondary signal portions, wherein said secondary encoders receive input from the first signal portion and are capable of providing encoded data regarding the first signal portion; and

(c) a data assembler for combining encoded data from said first encoder and said secondary encoders into an output data stream.

2. (Original) The system of claim 1 further comprising a decoder for reconstructing the input signal from information in said first signal portion.

3. (Original) The system of claim 1 wherein dividing the input signal is done in the frequency domain, and the first signal portion corresponds to the base band of the input signal.

4. (Original) The system of claim 1 wherein said signal portions are encoded at sampling rates different from that of the input signal.

5. (Original) The system of claim 2 further comprising one or more secondary decoders for decoding information encoded by said secondary encoders.

6. (Original) The system of claim 1 wherein said first encoder and said secondary encoders are embedded encoders.

7. (Original) The system of claim 1 wherein said splitter is a filter bank.

8. (Original) The system of claim 1 wherein said splitter is a Fast Fourier Transform (FFT) computing device.

9. (Original) The system of claim 8 wherein said splitter divides the input signal into M octave bands.

10. (Original) The system of claim 9 further comprising M1 decoders, $1 \leq M1 \leq M$, for providing an output signal that reconstructs the input signal from information in M1 signal portions of the input signal.

11. (Original) The system of claim 10 wherein the output signal has sampling frequency that is 2^{M1} times lower than the sampling frequency of the input signal.

12. (Original) The system of claim 1 wherein said output data stream comprises data packets suitable for transmission over a packet-switched network.

13. (Original) The system of claim 12 wherein said data packets are prioritized in accordance with the signal portion they represent.

14. (Original) The system of claim 12 wherein said data packets are assembled as to represent said two or more signal portions of the input signal.

15. (Original) A method for processing audio signals comprising:

(a) dividing an input audio signal into a first and one or more secondary signal portions, which in combination provide a complete representation of the input signal, wherein a first signal portion contains information sufficient to reconstruct a representation of the input signal;

(b) providing first encoded data about the first signal portion, and secondary encoded data about at least one secondary signal portion, wherein said secondary encoded data further comprises information about the first signal portion; and

(c) combining said first encoded data and said secondary encoded data into an output data stream.

16. (Original) The method of claim 15 further comprising the step of decoding the output data stream to reconstruct the input signal.

17. (Original) The method of claim 15 wherein said signal portions are encoded at sampling rates different from that of the input signal.

18. (Original) The method of claim 15 wherein said dividing is performed as a Fast Fourier Transform (FFT) computation.

19. (Original) The method of claim 18 further comprising the step of decoding the output data stream using M1 decoders, $1 \leq M1 \leq M$, for providing an output signal that reconstructs the input signal from information in M1 signal portions of the input signal.

20. (Original) The method of claim 19 wherein the output signal has sampling frequency that is 2^{M1} times lower than the sampling frequency of the input signal.